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L4-DMF: An Enhanced Self Adaptive Congestion Control Mechanism in Layer-4, a Simulation Validation

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Abstract— As part of an ongoing research, this work presents an enhanced self adaptive congestion control mechanism in layer-4, called Dynamic Modulation Feedback (DMF) which can fit into the network condition dynamically according to the parameters given by upper layer and the network layers These layers comprise of buffer windows, data rates, Average Queuing Length (AVQL), channel conditions or Packet Error Rate-PER. In this work, an optimization model for throughput maximization in Flow Aware-Wireless Local Area Network (FA-WLAN) was developed. A simulation was carried out with OPNET IT Guru to compare the congestion control mechanism of TCP- Tahoe, Reno, New Reno, Vegas and Sack (TCP-TRONVS) with a proposed layer-4 TCP-DMF. Performance metrics such as Latency, Throughput, Buffer Utilization, and Packet Loss Ratio, was used in the analysis while the measurement results from OPNET Modeler 9.1 was analysed. Consequently, this work showed that with the proposed layer-4 DMF congestion control mechanism, FA-WLANs will scale gracefully in this new era of High Performance Computing (HPC).

Keywords — Channel, Congestion, Layer-4 DMF, Optimization, Packet Loss Ratio, Performance, TCP-TRONVS

1. INTRODUCTION

CONGESTION management in wired and wireless environments has continued to attract attention in the networking market segments. The Internet topology is changing fast, and more often than not includes a Wireless last Hop. The number of internet services increases rapidly every year like TCP Hypertext Transmission Protocol (HTTP), Domain Name Service (DNS), Hypertext Transmission Protocol Secured (HTTPS), etc. A user has a wide range of possibilities, not only in choosing the applications, but also in selecting the end user devices or the connection method [1]. Today, packets may get lost on wireless links, for instance due to radio interferences.

This is a dramatic change over the assumption that was made in wired networks, where most (if not all) of packet losses were due to network congestion or buffer overflow [2]. While the congestion control mechanisms of TCP-TRONVS offers a good stability for inelastic applications and traffic flows, adapting WLAN in high performance computing requires an extensive study of its congestion management schemes while proposing a possible enhancements.

In data networking andqueuing theory, network congestion occurs when a link or node is carrying so much data that its quality of service deteriorates. Typical effects include queuing delay, packet loss or the blocking of new connections. A consequence of these latter two is that incremental increases in offered load lead either only to small increase in networkthroughput, or to an actual reduction in network throughput [3].

Network protocols which use aggressive retransmissions to compensate for packet loss tend to keep systems in a state of network congestion even after the initial load has been reduced to a level which would not normally have induced network congestion. Thus, networks using these protocols can exhibit two stable states under the same level of load. The stable state with low throughput is known as congestion collapse [3] while we refer the stable state with high throughput as congestion upgrade. This work defines WLAN congestion collapse as a condition which a packet switched Access Point (AP) halts traffic flow to congestion. Congestion collapse generally occurs at choke points in the network, where the total incoming traffic to a node exceeds the outgoing bandwidth. In a heterogeneous link, connection points between a local area network and a wide area network are the most likely the congested points. When a network is in such a condition, it settles (under overload) into a saturation stable state where TCP traffic demand is high but little useful throughput is available, and there are high levels of packet delay and loss (caused by routers discarding packets because their output queues are too full) and general quality of service is extremely poor.

Generally, Transmission Control Protocol (TCP) which utilizes a network congestion avoidance algorithm that includes various aspects of additive increase/multiplicative decrease (AIMD) schemes, with other schemes such as slow-start in order to achieve congestion avoidance. The TCP congestion avoidance algorithm is the primary basis for congestion control in the Internet [4], [5]. To avoidcongestion collapse, TCP uses a multi-faceted congestion control strategy to handle layer-4 issues which is not ideal. However, TCP-TRONVS is very sensitive to packet losses and requires further improvements to better adapt to the wireless environments. A good congestion control scheme should be able to dynamically handle packet arrivals at the network edges, hence allowing fair sharing of resource among all the users leading to a significant bandwidth optimization.

1.1. Our Contribution

Congestion control is the problem of managing network traffic or a network state where the total demand for resources such as bandwidth among the competing users exceeds the available capacity. This work views it as a layer-4 problem in the OSI reference model and it is a core infrastructural problem stemming from the packet switched and statistically multiplexed nature of the Internet and has an impact on the Internet stability and manageability for realistic loads. Hence, this work proposed an improved congestion management scheme called TCP layer-4 Dynamic Modulation Feed (L4-DMF) which will address both packet losses even at busty traffic and congestion in a FA-WLAN.

2. RELATED WORKS

The Flow-Aware Networking concept, first proposed in 2004 [6] is relatively new. The main goal of FAN is to achieve efficient packet transmission with the minimal knowledge of the network. In FAN, the traffic is sent as flows (streaming or elastic). The first type of flows is usually used by real-time applications while the second one carries best effort traffic. Two scheduling algorithms were proposed for FAN, PFQ (Priority Fair Queuing) [7] and PDRR (Priority Deficit Round Robin) [8]. Another proposal for realizing the FAN concept, called AFAN (Approximate FAN), was described in details in [9]. These FAN versions yield similar results. That is why the simulation analysis presented in this paper is provided only for the PFQ algorithm. FAN is a scalable solution. The work [10] opines that the complexity of the queuing algorithms does not increase with the link capacity. Moreover, fair queuing is feasible, as long as the link load is not allowed to attain saturation levels (it is ensured by the admission control).
According to [11], four congestion control mechanisms have been proposed for Flow aware Networks, FAN i.e., EFM (Enhanced Flushing Mechanism), RAEF (Remove Active Elastic Flows), RBAEF (Remove and Block Active Elastic Flows), and RPAEF (Remove and Prioritize in access Active Elastic Flows).
The work in [12] argues that TCP congestion avoidance algorithm is the primary basis for congestion control in the Internet [12]. The work in [13] used simulations to explore the benefits of adding selective acknowledgments (SACK) and selective repeat to TCP. It compared Tahoe and Reno TCP, the two most common reference implementations for TCP, with two modified versions of Reno TCP. The first version is New-Reno TCP, a modified version of TCP without SACK that avoids some of Reno TCP's performance problems when multiple packets are dropped from a window of data. The second version is SACK TCP, a conservative extension of Reno TCP modified to use the SACK option being proposed in the Internet Engineering Task Force (IETF). The work then described the congestion control algorithms in a simulated implementation of SACK TCP while showing that selective acknowledgments are not required to solve Reno TCP's performance problems when multiple packets are dropped. Again the absence of selective acknowledgments does impose limits to TCP's ultimate performance. The work concluded that without selective acknowledgments, TCP implementations will only be constrained to either retransmit at most one dropped packet per round-trip time, or to retransmit packets that might have already been successfully delivered. The congestion control and packet retransmission algorithms in Tahoe, Reno, New-Reno, and SACK TCP will be presented in the next section.
The author in [14] presented a study on congestion control and scheduling in communication networks. In contrast to standard protocol design where there is minimal communication between the scheduling and the congestion control, the paper argued that there are a number of benefits to jointly optimizing these algorithms, especially in wireless networks. The first, the benefit of the coordination is that effective buffer sizing, is achieved even when channel rates are variable in the WLAN. The second benefit from [14] is that we can prevent conflict situations where the congestion control and the scheduler both try to assign bandwidth to the flows. The third benefit is that coordination allows us to prove theoretical utility maximization results that are not affected by possible oscillations.
In analysis done by Kelly, Maulloo and Tan in [15] it was shown that many variants of TCP can be viewed as approximations of primal-dual algorithms that solve an underlying optimization problem. However, in most cases as studied in literature, this analysis abstracts away some of the details of the scheduling problem. From this research, it was observed that congestion control mechanism implemented through Queue Management algorithms, is considered a key factor to solve this optimization problem above while keeping TCP/IP networks efficient and reliable from the user's viewpoint.
Many works in literature have investigated methods to estimate the key wireless link parameters in the context of congestion metrics, for example, link bandwidth estimation [16], link buffer size estimation such as max-min, loss-pair and sum-of delays [17], [18], and queue length estimation [19]. For this reason, the utilization of fuzzy logic (FL) has shown to be useful in designing new active queue management (AQM) methods [20] that can be used to alleviate congestions in wired and wireless networks. For example, Mallapur *et al.* [21] proposed a buffer manager located at the base station using a fuzzy controller for packet dropping in wireless cellular networks. The controller uses three fuzzy parameters, namely application priority, queue length and packet size.
Many researches have been devoted on the queue size management algorithms such as Random Early Detection (RED) and Weighted RED, PID controller [22], and NLRED to reduce the likelihood of global synchronization, as well as keeping queue sizes down in the face of heavy load and bursty traffic. Other related AQM schemes such as GRED, WRED, and ARED are summarized in [23]. The Random Early Detection (RED) mechanism can solve the Drop Tail's (DT) deficiencies well. It uses randomization to ensure that all connections encounter the same loss rate. By dropping packets before the router's buffers are completely exhausted, the RED mechanism try to prevent congestion. In [24], another approach known as Random Exponential Marking (REM) was also developed to measure the average queue size instead of congestion measure. It was able to achieve high link utilization, neg-

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ligible packet loss and short queuing delay in a simple scalar manner. TCP assumes that losses of packets are derived from a buffer overflow at a Base Station (BS) router. It invokes the congestion control to reduce traffic unnecessarily when the packet loss occurs due to transmission errors on a wireless link. Thus one approach to mitigate this problem and distinguish packet loss due to congestion and others packet errors due to channel variations, is to employ TCP proxy called Performance Enhancing Proxy (PEP) [25]. However, the major problem of such Indirect-TCP (e.g. Snoop protocol) and M-TCP is a hand-off latency due to mobility of (mobile host) MH [26], [27].

The observed limitations of the various congestion management schemes studied in sample literatures is outlined as follows:

1. The quantitative evaluation of the performance of TCP over 802.11 WLANs shows that TCP congestion control schemes cannot handle packet losses effectively in a FA-WLAN for high performance computing.
2. There is no discussion on real physical parameters of the realistic loads and feasible network parameters for congestion control in FA-WLAN.
3. The congestion management schemes owing to the structure of their algorithms introduce latency, thereby affecting the traffic throughput.
4. There is no dynamic provisioning on the network edge devices such as APs to manage queues regardless of the data rates.
5. There is unfairness in throughput delivery as the throughput for nodes depends on the number of flows that it has (TCP oriented).
6. Active Queue Management schemes focus more on packet drops on exceeding the equilibrium threshold rather than packet fragmentation or implicit feedback (such DMF)

Consequently, a FA-WLAN model for high performance computing must demonstrate its capability in addressing these limitations as observed in context.

3. METHODOLOGY

3.1. FA-WLAN Congestion Management

As discussed in our earlier work, the observed limitations of TCP-TRONVS in IEEE 802.11b Wireless Networks gave rise to the proposed TCP layer 4-DMF. A fuzzy logic implementation in a rule editor of MATLAB 2009b which is referred to as Fuzzy-Logic Adaptive Queuing controller (FLAQ) inherits from the classical Random Early Detection (RED) algorithm. TCP layer 4-DMF has an implicit dynamic modulation feedback. As a variant of TCP FLAQS, this was adapted in FA-WLAN for congestion analysis. As shown in Fig. 4.1, the fuzzy DMF controller predicts dynamically the packet dropping rate (joint scheduling and congestion control) and the corresponding average queue length. It relies on the average queue length at the base Access Point router (AP) and the packet loss rate caused by the channel variations in mobile environment; assuming there is no buffer overflow due to the congestion. Using this model, a heuristic TCP performance can be estimated over a time-varying channel under different conditions of user's mobility.

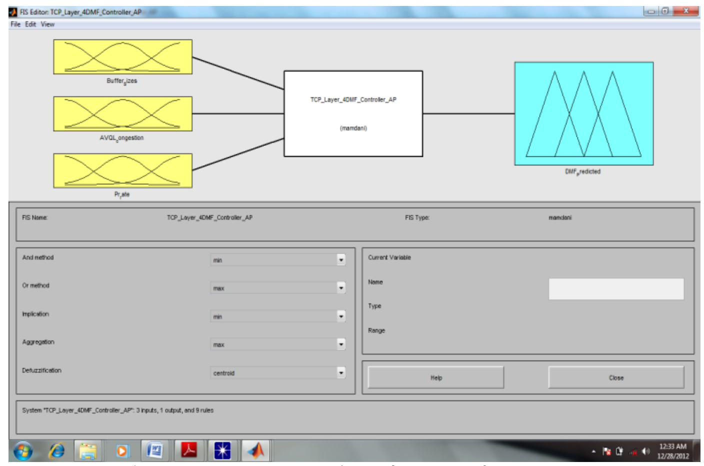

Fig.1: TCP layer-4DMF mamdani fuzzy inference engine

When the arriving packets cannot be accommodated due to lack of network resources (bandwidth, buffer size, etc), this indicates congestion occurring at router buffers of networks. More specifically, a poor network performance due to congestion can be offered in terms of high dropping and queuing delay for packets, low throughput and not maintaining the average queue length which may not prevent the router buffers from building up, then dropping packets.

However, congestion control mechanism implemented through intelligent Queue Management algorithms, is assumed to be the key factor to solve this problem keeping TCP/IP networks efficient and reliable from the user's viewpoint.

The result of the proposed layer 4-DMF showed a significant improvement in TCP QoS (throughput, etc) performance considering the user's mobility and realistic traffic loads in the FA-WLAN.

4. ANALYTICAL DESIGN

4.1. Management System Representation/Planning

Given a set of AP equipment and a set of available channels, the objective function is to design an efficient network plan for a FA-WLAN such that maximum wireless coverage of a specified area is achieved and that congestion in the wireless topology (BSS) is efficiently managed. In this context, the efficiency of a network plan E_f is defined by a channel overlap measure and the net throughput expected over the wireless service area for a single user in the FA-WLAN.

Let A be the set of candidate AP locations, and let M ($0 < M \leq |A|$) denote the maximum number of APs to be deployed and positioned. The specific locations are determined in advance with respect to factors such as potential installation costs, accessibility, physical security, radio propagation aspects, health safety, psychological factor, etc. Fig. 4.2 shows

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the AP LTP used in this research.

The active service area is represented by a grid of Localized Test points (LTPs) with a given resolution. Let *J* denote the set of LTPs to be covered. The received signal strength at LTP *j* depends on the transmission power level *TPL* (AP_i) at the serving AP_i and attenuation At_n between the AP and LTP *j*. The received signal strength be computed as:

$$Rss\, j^{(rec)}: \leftarrow At_n.TPL\,(AP_j) \quad (1)$$

The transmission power TP_j (AP_j) is assumed to be fixed and ascertained in advance. The attenuation At_n is given in a path-loss prediction grid in channel model shown in Fig.1

4.2. FA-WLAN Design Assessment

4.2.1. Coverage

A LTP *j* (*j€J*) is covered if there is at least one installed AP_j (*j€A*) such that the received signal from *j* satisfies $RSS_j^{(rec)} \geq Ø^{(r)}$, *where* $Ø^{(r)}$ is the receive sensitivity threshold (RST). This parameter $Ø^{(r)}$ defines the minimal signal strength required for receiving transmissions at the lowest possible data rate. This threshold can be adjusted as it is an adjustable configuration parameter with its typical value found in the APs documentation manual. In this context, a network with very extensive coverage is desired in the FA-WLAN.

4.2.2. Throughput

To ascertain our throughput, OPNET IT Guru is used to find a fitting function that represents the throughput experienced by a user in a FA-WLAN environment. The throughput depends on the strength of the strongest signal received by any AP, hence it is assumed that the transmit power to be fixed on the attenuation value. Given that a client AP is denoted by *a*, then the net throughput of the user *j* is denoted by $Ø\,(TPL_j)$.

The throughput in some area is a strong indication that a better AP placement is needed. Again, our new objective function is to maximize the total throughput over all LTPs by choosing an appropriate subset of candidate AP locations.

4.3. Optimization Models

The goal in our FA-WLAN planning and design is to maximize the throughput that a user can expect regardless of the traffic type. Two aspects are considered viz:

a. Data rate (contention Quality at the physical layer)
b. Contention for the medium with other users.

The first on depends on AP locations ie a user experiences a higher throughput if the serving AP is closer to the user in terms of attenuation while in the second case, contention among users depends on the active users the serving APs and the channel assignment.

4.4. AP Placement/Localization for Channel Optimization in FA-WLAN

The AP placement determines the maximum data rate that a user can expect. With growing distance to the closet AP, the net data rate decreases as shown in Fig. 4.2. The max possible data rate is only achieved if a user does not have to contend with the medium. Under this assumption, maximizing the average data rate taken over all TPs allows the expected user's throughput. The goal in this context is to maximize the throughput that a user can expect. This leads to a problem formulation in integer programming model viz:

Given a typical facility location model, how can the problem of maximizing the expected throughput with M most installed AP be realized?

From the formulation above, in a typical facility location model in FA-WLAN, let us use two classes of binary variables vz: variables $z_a \epsilon\{0,1\}$ for all potential locations in a, and variables $z_{aj} \epsilon\{0,1\}$, for all pairs of locations and TPs. Here Za =1 suggests that an AP is installed at location a. Again, Xaj = 1 means that TPj is associated to APa. It is worthy to note that variable Xaj can be set to zero if the received signal strength in TPj from APa is below received sensitivity threshold $Ø^{(r)}$.

Our objective function in maximizing throughput in a congested network is given by:

$$Max \frac{1}{|j|}.\sum_{a.} j Ø(At_n)\, x_{aj} \quad (1)$$

Subject to:

$$x_{aj} \leq z_a \quad (2)$$

$$\sum_a x a_j \leq 1 \quad (3)$$

$$\sum_a z_a \leq M \quad (4)$$

$$Z\epsilon\{0,1\}^A,\ x\ \epsilon\ \{0,1\}^{A*J} \quad (5)$$

From equation (1), the objective function measures average throughput per TLP, constraint equation (2) states that a LTP can only be assigned to an AP if the AP is installed, (3) Ensures that each TP is assigned at most once, and equation (4) & (5) limits the number of APs.

In this work, since the existing methods lacks implicit feedback controls and cannot effectively handle congestion in FA-WLAN under realistic loads, this work therefore developed a layer-4 DMF based on Fuzzy-Logic Adaptive Queuing controller (FLAQ) with modulation feedback at the AP base station in order to tune the average queue length and the wireless packet error under realistic load conditions.

The proposed model seeks to maintain buffer size space over a time-varying channel which may exhibit significant degradation in the network bandwidth estimation. Then, the work predicts the lowest packet dropping received by the mobile receiver over a FA-WLAN when there is congestion effect at the AP base station buffer. By using this model, a heuristic TCP performance can be evaluated over a time-varying channel under different conditions of user's mobility and realistic load conditions. Fig. 4.2 shows a captured AP placement in LTP.

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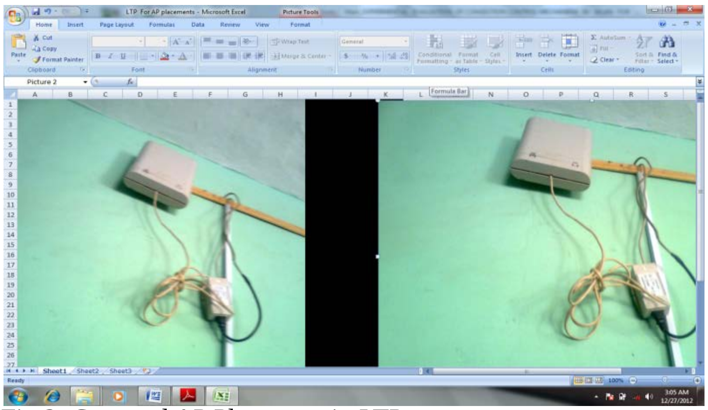

Fig.2: Captured AP Placement in LTP

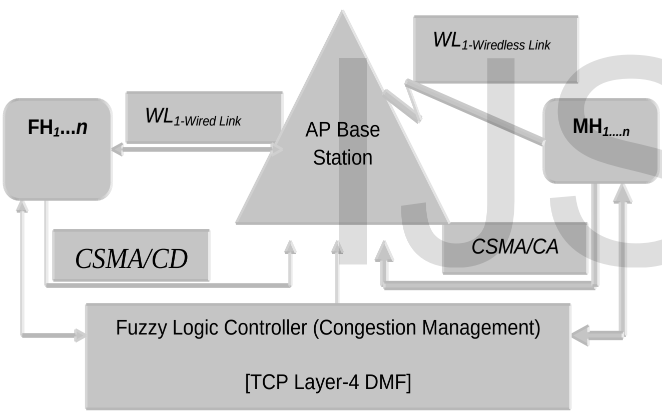

Fig 3a: Unicast-Broadcast FA-WLAN model

Fig.3b: FA-WLAN Process Model

Fig.3c: System Architecture for FA-WLAN

The proposed TCP layer 4 DMF scenario is shown in Fig. 3a. Fig. 3c shows the system architecture. From Fig. 3b, the considered key preliminaries required to predict the congestion (packet dropping rate, etc) based on our earlier work on fuzzy logic adaptive queuing controller in the FA-WLAN and Rayleigh fading channel is as follows:

- Multiple TCP flow traffic is considered for all mobile receivers.
- The network is assumed to be stable without heavy or bursty TCP traffic.
- The TCP packet error rate (PER), (i.e., *Pr*), is caused by the variations of wireless channel when only highly bit errors occurs during traffic transmission. Assuming there is no congestion at the router buffer of AP base station.
- *Pr is* measured by the channel estimator at mobile receiver and returned back via the ACK feedback of the round trip of TCP to indicate the sender about the channel bit errors, so we assumed Pr changes from 5% to 30%.
- The TCP rate regulator at the AP router queue of the AP base station is required *if and only if* multiple TCP flows are present. This rate regulator is mainly used to distinguish the packet error (dropping) due to the variations of wireless channel and the packet loss due to congestion of buffer overflow. In our assumption, there could be packet dropping due to AP buffer overflow is extreme congestion. So, the link could be under or over utilizing bandwidth and the queue threshold of the APs could be exceeded.

At the AP base station (BS), we considered the following assumptions:

- Let the buffer size = 256kb packets
- The AP router queue with *Qmin* = 50kb [packet], and *Qmax* =150kb [packet]
- If the average queue length is less than 50kb, no packets are dropped➔No TCP congestion
- If the average queue length is more than 150kb, all the arriving packets are queued while DMF regulates the feedback flows to and from the MH or MNs.
- If the average queue length is between *Qmin* and *Qmax,* then the packets should be controlled by the fuzzy logic controller depending on to inputs $(AVQL_{congestion}, P_r)$

Packet loss rate at AP base station is compensated by DMF

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algorithm (hence, there is no packet loss at the event of congestion) under realistic loads.

5. SYSTEM'S IMPLEMENTATION

5.1. Simulation Testbed

In this research, the validation of the proposed TCP Modulation feedback (L4-DMF) algorithm is considered very vital as this will help to improve congestion management scheme by addressing both packet losses even at busty traffic in a FA-WLAN. In this context, we compared the performance of the developed algorithm (proposed L4-DMF) with that of TCP-TRONVS while establishing the influence of *QoS* parameters in the developed testbed.

To demonstrate more details on the proposed TCP Modulation feedback (L4-DMF) algorithm implementation, a simulation testbed was built with OPNET IT Guru 9.1 with DMF configured in the OPNET engine as shown in Fig. 4.19, that consists of a two wired LAN PC servers (HTTP and FTP/WEB servers), 40mobile client nodes, two AP base stations and an IP gateway router, all connected to the IP cloud. In the mobile nodes including the APs, TCP Tahoe, TCP Reno, TCP NewReno, TCP SACK, TCP Vegas as well as the TCP layer-4 Modulation feedback (DMF) algorithm were all configure in six independent scenarios. The subnet sites and IP gateway cloud emulates the FA-WAN link with the desired throughput, latency, buffer utilization, data rate stability, queuing length and packet loss rate.

5.2. Simulation Configurations

This work implemented Fig. 3c in OPNET IT Guru which was used to generate the parameters for various case scenarios in the simulations. Traffic attributes for the FA-WLAN are listed in Table 4.2. The runtime environment attribute were as follows in the OPNET simulator.

Table 4.2: Basic OPNET Traffic Attributes

	Configurations	Values
1.	Simulation Duration for Each Scenario	120 mins.
2.	Link Propagation Delays	0.5µsecs
3.	Switch Output Buffer	100 packets
4.	Simulation Seed	128
5.	Update Interval	50000 Events
6.	Simulation Kernel	Optimized
7	TCP Variants	Configured
8	Mobile clients(Max-min)	[40-5]

Running the test bed, we measured the above metrics achieved by the six TCP versions. Fig. 4.19 shows the validation testbed while Fig 4.20 shows the simulation run/compilation.

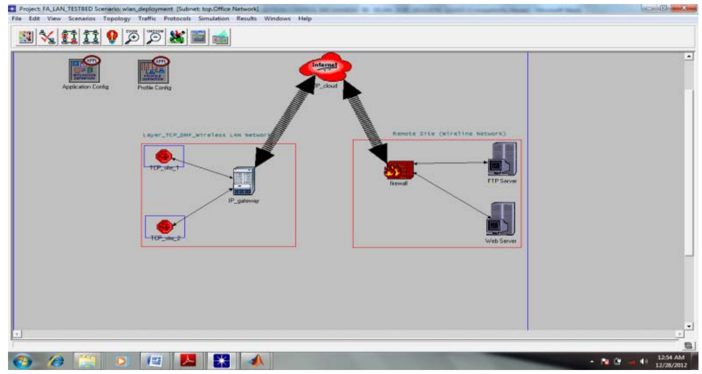

Fig.4.19: Validation testbed with FA-WLAN sites_1 and site_2

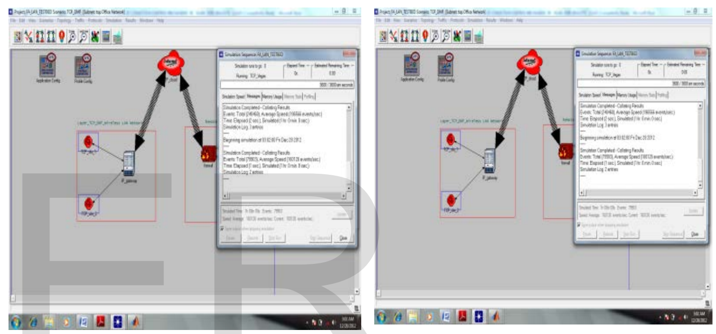

Fig.4.20: Simulation Plot of TCP-TRONVs with the proposed layer-4 DMF algorithms

5.3. Results and Discussions

Latency Response: In this work, an evaluation on various TCP variants including the TCP SACK was carried out via the simulation model. Fig. 4.21 showed the latency plot of TCP-TRONVs. TCP Vegas and the proposed TCP DMF show a similarity latency response under realistic loads. The average latency for all TCP variants except Vegas and the proposed TCP DMF maintained a fast rise latency throughout the transitions as depicted by the trend curve beginning from 0.00125ms up to 0.00127ms for TCP Tahoe, Reno, NewReno and SACK. In the same vein, the proposed TCP DMF showed a comparative latency response with TCP Vegas (0.0075ms). It maintained a steady rate of about 0.0076ms relative to TCP Vegas. Therefore a difference in the values of latency in the face of equal load per time over the AP controller makes the frame size distribution over TCP variants a major consideration.

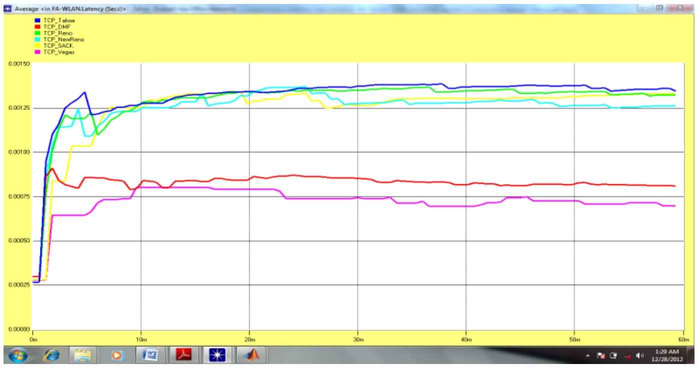

Fig.4.21: latency Plot of TCP-TRONVs with the proposed layer-4 DMF algorithm

Throughput Response: Fig. 4.22 compares the steady-state throughput response achieved by all TCP variants. Interestingly, it was observed that the proposed layer-4 TCP DMF algorithm had slightly better throughput behavior of about 8870 packet/bits compared with TCP Tahoe, Reno, New Reno, SACK and Vegas. This is due to its connection oriented behavior leading to accurate fuzzy estimate of the available bandwidth, buffer size, average queuing length, packet error rate, and fair distribution of frame sizes (packets) under realistic load conditions. As a result of this better throughput behavior, the transmission of realistic traffic witnessed reliable frame data delivery with active connections transmitting data between the mobile nodes and the AP server, with an emulated round trip time equal to 100 ms (a near zero packet loss rates).

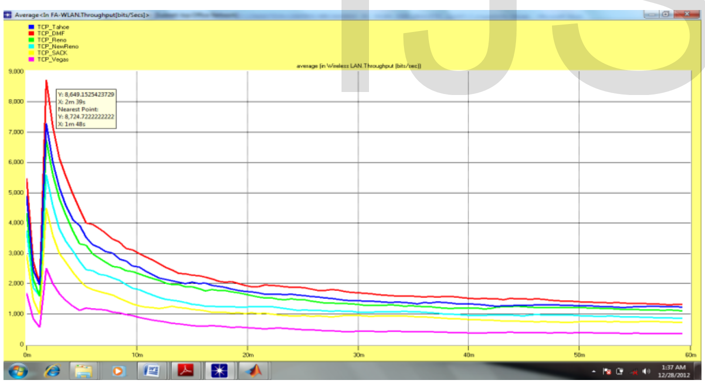

Fig. 4.22: A throughput Plot of TCP-TRONVs with the proposed layer-4 DMF algorithm

Buffer Utilization Response: Fig. 4.23 shows the buffer utilization plot of TCP-TRONVs with the proposed layer-4 DMF algorithm. At the core AP base station and mobile stations, the output buffer was set to hold a maximum of 2048Kbps bytes. Essentially, the buffer sizes were varied starting from 64kb packets to 2028 packets for various realistic load sources. It was observed that with the proposed layer-4 DMF algorithm, there was a fairly assumed peak exponential rise in buffer utilization with time due to congestive load effects. Traffic flows shows an enhanced throughput as the buffer sizes were increased. The buffer utilization curves are on top of each other with gradual transitions giving way to higher and efficient throughputs. From a buffer size of about 2000kb packets and above, throughput of bulk traffic TCP DMF was significantly high. This guarantees a fair allocation in the congested link. Thus, an increase in buffer size leads to a better performance network considering the various load intensities. This also means that a better buffer size utilization accounts for less packet drops in a congestive wireless link.

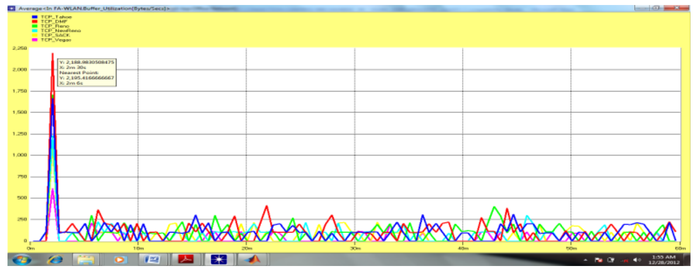

Fig.4.23: Buffer utilization Plot of TCP-TRONVs with the proposed layer-4 DMF algorithm

Packet Loss Ratio: Fig. 4.24 depicts a plot of Packet Loss Ratio of TCP-TRONVs with the proposed layer-4 DMF algorithm. As observed from the plot, the proposed layer-4 DMF algorithm has a relatively good packet loss ratio as a result of its enhanced throughput behavior while TCP Vegas which has a comparatively better Packet Loss Ratio behavior under congestive scenario. Ideally, the TCP Vegas is demonstrates a higher PLR while TCP Tahoe depicts a weak PLR.

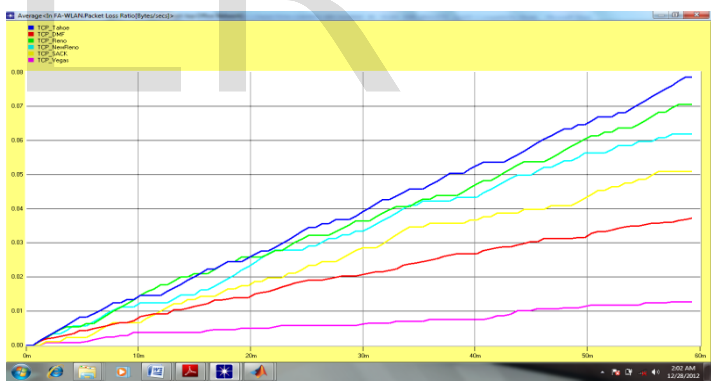

Fig. 4.24: Packet Loss Ratio Plot of TCP-TRONVs with the proposed layer-4 DMF algorithm.

Average Queuing length: The proposed TCP_DMF has better Queue management owing its intelligent AVQL prediction. Fig. 4.25 shows the Queuing Length plot of TCP-TRONVs with the proposed layer-4 DMF algorithm. As depicted, the queuing variations with the proposed algorithm show a better spread and congestion control owing to the active queue management (AQM). The TCP Tahoe, Reno, NewReno and SACK on the other hand shows a better management in regions of little or moderate congestion while Vegas and the proposed DMF shows better AVQL management under intense congestion considering the preset equilibrium thresholds.

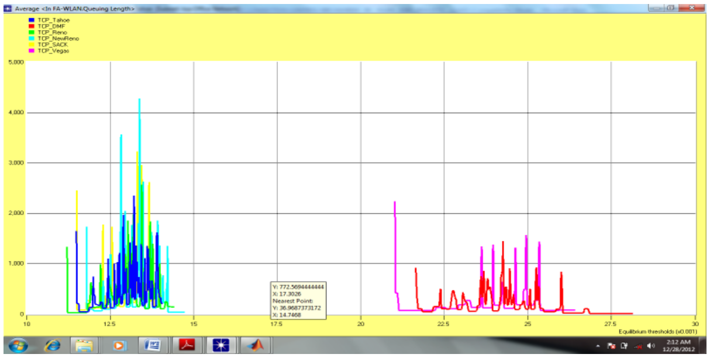

Fig. 4.25: Queuing Length Plot of TCP-TRONVs with the proposed layer-4 DMF algorithm

CONCLUSIONS

This research shows that TCP-TRONVS are often unable to give accurate estimates in all FA-WLAN metrics, thereby making it as a congestion scheme unsuitable for future TCP cloud services. Although there is still room for improvement in TCP performance, the main factor limiting all TCP performance is random packet loss. To overcome this problem, this work propose a new algorithm, layer-4 DMF, that enables TCP to achieve both a higher throughput over wireless links and fair behavior over wired links. The performance of this new algorithm has been compared with that of other existing TCP, and it has been found that significant improvement is obtained using the proposed layer-4 DMF taking cognizance the metrics such as Latency, Throughput, Buffer Utilization, and Packet Loss Ratio.

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